MicroDAC IV

PORTABLE PROGRAMMABLE DIGITAL FILTER

USER'S MANUAL

DIGITAL AUDIO CORPORATION

A DRI COMPANY

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User's Manual

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1.0 OVERVIEW

1.1 Introduction

The MicroDAC IV is a portable, programmable digital filter targeted for applications requiring unattended filter deployment. A replacement for the venerable MicroDAC III, this powerful signal processor is fully self-contained and features:

- Easy setup and operation
- Operation with live voice signals, cassette and microcassette tape recorders, video cassette recorders, telephone taps, and radio receivers
- 12VDC power input enables operation from AC power (external adaptor included) or from 12V automotive battery system
- Compact enclosure (1.5” H x 7.3”W x 10.0”D)
- Fixed bandwidth of 5.4kHz
- Three factory preset adaptive filtering modes
- Four programmable filtering modes

Key Features

- Stereo processing capability, in addition to standard monaural
- Windows™ programming software, which allows easy, intuitive reprogramming of filters for specific applications
- Crash Detect, which will automatically clear the adaptive filter should it crash
- Reference Compensation, which automatically adjusts the **RIGHT (REF)** input level as necessary to avoid coefficient saturation and maintain optimum cancellation
- Auto Mu, which maintains adaptive filter stability by automatically scaling adapt rate for both input signal power and filter size
- Both standard LMS (Least Mean Square) and LAV (Least Absolute Value) adaptation algorithms available
- EXTERNAL CLEAR connector, which allows manual clearing of the adaptive filter by remote switch closure
- Dynamically partitionable filter array, which allows combining large and small filters in series
- Input Limiter, which automatically reduces input levels on overload
- AGC, which automatically boosts level of processed signal prior to output
- Digital Output (S/PDIF format) for recording directly to DAT

2.0 Minimum Hardware Requirements

NOTE: Computer is required only for programming the MicroDAC IV; it is not required for stand-alone operation.

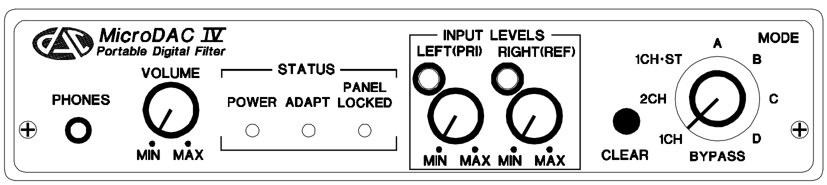
- 100 MHz CPU (Pentium or equivalent)
- Windows95/98/NT operating system
- 8 Megabytes of RAM
- At least 4 Megabytes spare hard disk space
- 640x480 color VGA display
- Two button mouse
- At least one spare RS232 port

2.0 OPERATION

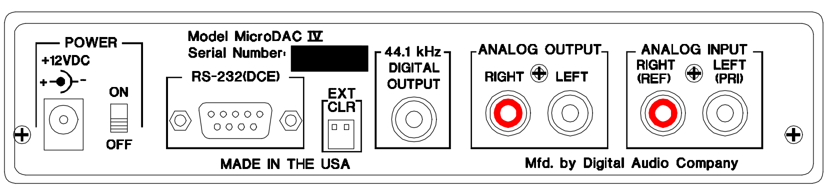
MicroDAC IV promotes ease-of-use by reducing the number of buttons and switches the user needs to learn. All of the settings that required adjusting internal DIP switches in the previous version can now be accomplished using a computer. The user configures the filter(s) as desired using special Windows software, then downloads the settings to the MicroDAC IV where they are stored in non-volatile memory.

The following section discusses the connections and controls for the unit. The remaining sections describe typical setups and provide an overview of adaptive filters to ease the selection of the best filter in a particular situation.

2.1 Controls and Connectors



Front Panel View



Back Panel View

Figure 1: MicroDAC IV Panels

Refer to Figure 1 for an illustration of the front panel features, described as follows:

- Standard 1/8” stereo headphones may be plugged into the **PHONES** connector. Adjust loudness with the **VOLUME** control beside the connector.

- The green **POWER** LED indicates that the MicroDAC IV is operating.

- The green **ADAPT** LED indicates that an adaptive filter is continuously updating its filter solution.

- The green **PANEL LOCKED** LED indicates that one or more of the front panel controls is locked and can only be changed by the Windows software.

- The **INPUT LEVEL** rotary knobs adjust the input signal levels on the LEFT (PRI) and RIGHT (REF) channels. Above each knob is a tricolor Level LED that indicates input signal strength. Adjust the LEFT and RIGHT **INPUT** controls so that the tricolor **LEVEL** LEDs display green or occasionally yellow. Do not allow the input level to go into the red region or distortion may occur.

- The **CLEAR** button is used to restart the adaptive filters as deemed necessary. This function is particularly useful if the filter has "crashed" and produces only garbled noise on its outputs (an extremely rare event when the Crash Detect feature is enabled). Another example of when to press the **CLEAR** switch is when the audio scene changes suddenly, requiring a new filter solution; this may occur when the microphone is moved, or when processing tape recorded conversations that occurred at different locations.

- The **MODE** rotary switch selects the desired filter by rotating the knob to the desired filter. The MicroDAC IV will automatically reset and load the filter whenever the MODE setting is changed. Notice that one of the filter settings is labeled **BYPASS**; this setting simply routes the input signal to the output without any processing.

Now refer to the Figure 1 illustration of the rear panel features, described as follows:

- The **POWER** connector mates to a 2.1 mm barrel plug. To power the unit connect the external AC power adapter to the **POWER** connector. Alternatively, 12VDC power can be directly applied using a specially-wired adaptor; the wiring guide for the plug is silk-screened on the rear panel.

- The MicroDAC IV communicates with the computer's serial port via the **RS232** connector. The supplied Windows software programs the MicroDAC IV over this serial link.

- An **EXTERNAL CLEAR** jack can be used to remotely clear the adaptive filters via contact closure (shorting the two terminals together). This operates in an identical manner to the front panel **CLEAR** button.

- The 44.1KHz **DIGITAL OUTPUT** is an S/PDIF format digital data stream that is driven directly from the signal processor output. This data is clocked at the same rate as CD players, which allows the digital processed signal to be recorded directly to a DAT or CD recorder.

- The **ANALOG OUTPUT** RCA jacks provide the MicroDAC IV's filtered output in analog form. These line level signals can be directly connected to a copying tape recorder or loud speaker amplifier, if desired. If the MicroDAC IV is not powered, the **ANALOG INPUT** signals are automatically routed to the **ANALOG OUTPUT** jacks.

- The **ANALOG INPUT** RCA jacks are the MicroDAC IV's inputs. These line level inputs allow the audio source to be input for filtering.

2.2 Fast Start

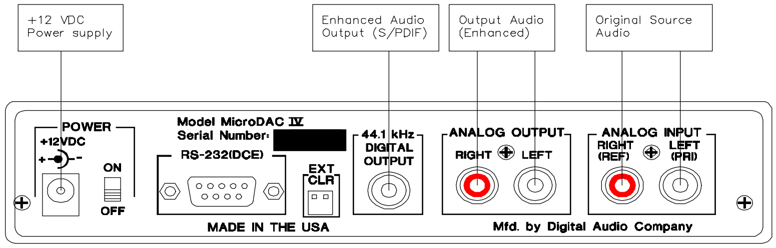


Figure 2: Basic Enhancement Setup

Fast start the MicroDAC IV as follows using Figure 2 as a guide:

1. Connect the audio source(s) to the **LEFT (PRI)** and **RIGHT (REF) ANALOG INPUT** RCA jacks on the MicroDAC IV rear panel as shown in Table 1. Although this table assumes a live microphone preamp connected to the input of the MicroDAC IV, the signal could come from many types of sources such as a cassette or micro-cassette recorder, DAT recorder, videocassette recorder, telephone tap, or radio receiver. Note that the **RIGHT (REF)** signal is only used in stereo configuration and/or with the 2CH Adaptive filter.

Table 1: MicroDAC IV Input Connections

MODE	LEFT (PRI) Input	RIGHT (REF) Input
1CH	Preamp output	Unused
2CH	Preamp output	TV/radio output
1CH STereo	Left preamp output	Right preamp output
A-D (Mono configurations)	Preamp output	TV/radio output, if 2CH Adaptive filter programmed; unused otherwise
A-D (Stereo configurations)	Left preamp output	Right preamp output

2. If you wish to record the enhanced audio, connect the line-level audio inputs (TAPE IN jacks) of your Enhanced Tape Recorder to the **LEFT** and **RIGHT ANALOG OUTPUT** jacks on the MicroDAC IV rear panel as shown. Alternatively, the digital output stream may be directly recorded by connecting the **44.1KHz DIGITAL OUTPUT** to the digital input of the DAT recorder.

3. Turn the phones **VOLUME** control to **MIN,** then connect the stereo headphones to the **PHONES** jack.

4. Rotate both **INPUT LEVEL** knobs located on the front panel fully counter-clockwise and power the MicroDAC IV on. Switch the front panel **MODE** switch to **BYPASS**, so the signal input will flow directly to the output without being altered. As the source audio (live or recorded) plays, adjust the front panel **INPUT LEVEL** knobs clockwise until the tricolor **INPUT LEVEL** LEDs flash

green. If these LEDs flash in the red color region, then the knob has been turned too far; reduce the corresponding **INPUT LEVEL** knob setting. Now adjust the **PHONES** knob to a comfortable listening level.

NOTE: If the 2CH Adaptive filter is used, it is recommended that the **RIGHT (REF)** input be set to a slightly higher level than the **LEFT (PRI)** input to ensure maximum cancellation.

5. Rotate the **MODE** switch to the desired filter position.

6. Start the recorder if necessary, and monitor the recording to ensure quality.

3.0 STANDARD FILTER CONFIGURATIONS

3.1 Selecting the Correct Filter Mode

The filters in the standard MicroDAC IV are optimized for noise cancellation to increase voice intelligibility. Simple noises (e.g., tones, hum) are generally greatly attenuated. However, more complex noises (e.g., ambient restaurant noise, bar noise) may not be as effectively attenuated, but should be somewhat reduced.

The **MODE** switch setting determines the type of filtering based upon the configuration programmed into the MicroDAC IV. When in the **BYPASS** mode, the MicroDAC IV simply transmits any audio applied to the **ANALOG INPUT** connector directly through to the **ANALOG OUTPUT** connector, bypassing the audio filters.

The three standard filter configurations correspond to the first three positions of the **MODE** switch; the remaining positions are programmable by the Windows software. The standard filters are one-channel mono (**1CH**), two-channel mono (**2CH**), amd one-channel stereo (**1CH·ST**).

The **1CH** filter does an excellent job of reducing simple time-correlated noises, such as tones, hums, and buzzes from a monaural signal. It also does an excellent job of reducing echoes and reverberations. Similarly, the **1CH·ST** filter removes tones, hums, buzzes, and echoes from a stereo signal, except that the filters' capability is spread across two signals, versus just one in the **1CH** case.

The **2CH** filter is used primarily for live radio/TV removal. If the incoming audio from a microphone is corrupted by interfering audio from a radio or television, a second radio/TV tuned to the same channel can be used as a cancellation reference for reducing the interfering audio and revealing the desired conversation. Simply connect the audio output from the second radio/TV to the **RIGHT (REF)** input, adjust the levels, and the cancellation will be performed automatically.

3.2 Hints For Effectively Using The MicroDAC IV's Filtering Capabilities

First, it is vital that the microphone installation and/or source material be prepared as carefully as possible. For example, if the source audio is from a tape recorder and the microphone was placed beside a loud air conditioner and the target voices are 40 feet away, it is doubtful any recoverable voice was recorded onto the tape, especially if the recorder used an automatic level control mechanism (While this sounds like an unusual example, it does happen frequently.) Also, make sure the heads are clean and properly aligned on the playback machine, to allow maximum recorded signal to be available to the signal processor.

Second, keep the input levels in the green to yellow zone at all times. Increasing the input level into the red zone could create distortion that will hinder the MicroDAC IV's filtering capabilities.

Third, if the filtered output of the MicroDAC IV is to be recorded, make sure that the recording machine's heads are clean and that the mechanism is properly lubricated and in good working order.

Finally, always use good quality, fresh tapes for all recordings, and if possible, monitor the recording as it is being made.

3.3 Factory Preset Filter Descriptions

1CH MODE:

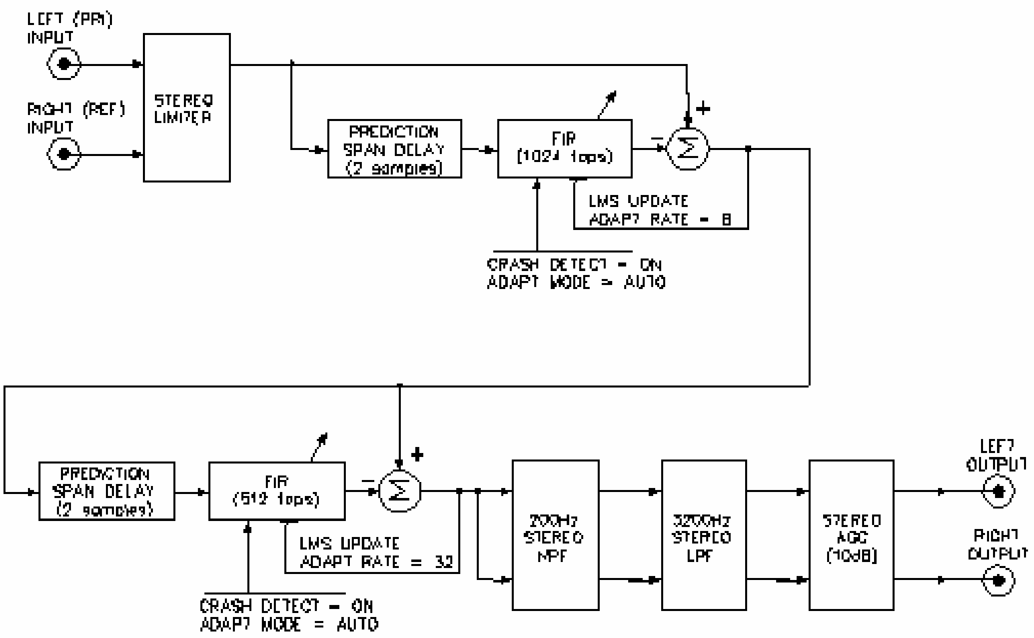


Figure 3: 1CH MODE Block Diagram

Notice that in **1CH** MODE, only the **LEFT (PRI)** input gets processed. This signal is first fed through the **Limiter,** which helps prevent the signal from distorting on overload.

The signal then enters the first stage 1CH Adaptive filter, which is configured as follows:

Table 2: 1CH MODE First Stage Factory Presets

Parameter	Preset Value
Filter Size	1024 taps
Prediction Span Delay	2 samples
Adapt Rate	8
Adapt Mode	Auto
Output	Residue
Algorithm	LMS
Crash Detect	Enabled

After the first stage of filtering, the signal enters the second stage 1CH Adaptive filter, which is configured as follows:

Table 3: 1CH MODE Second Stage Factory Presets

Parameter	Preset Value
Filter Size	512 taps
Prediction Span Delay	2 samples
Adapt Rate	32
Adapt Mode	Auto
Output	Residue
Algorithm	LMS
Crash Detect	Enabled

Next, the signal is 200Hz Highpass filtered, to remove any low frequency noises that may remain after processing, and 3200Hz Lowpass filtered to remove high frequency hiss.

Finally, a 10dB AGC is applied to correct for near party / far party talkers, and to provide a good output level for recording.

2CH MODE:

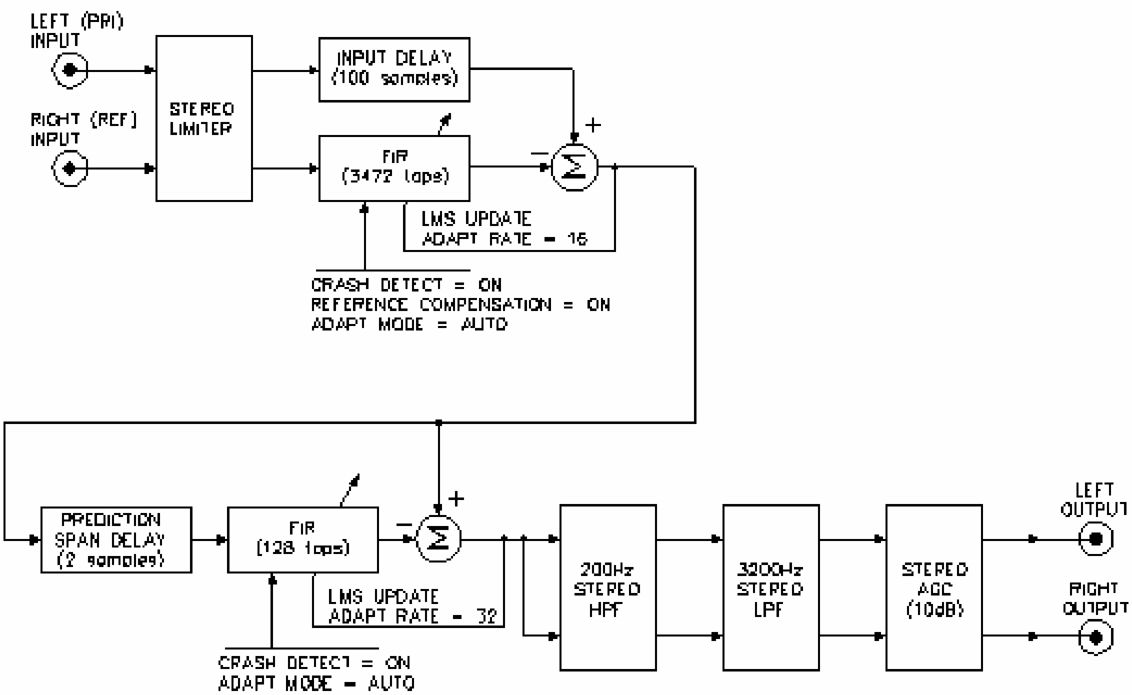


Figure 4: 2CH MODE Block Diagram

Notice that in **2CH** MODE, both the **LEFT (PRI)** and **RIGHT (REF)** inputs get fed into the processor. The **LEFT (PRI)** input is the signal from which interfering audio is to be removed, usually originating from a concealed microphone that has interfering radio/TV audio. The **RIGHT (REF)** input is the noise reference, usually the audio output from a second radio/TV tuned to the same channel.

The signals are first fed through the stereo **Limiter,** which helps prevent the signals from distorting on overload. The **Limiter** internally "links" the two channels, such that an overload on either input channel will cause the levels to be reduced equally on both channels. This minimizes the impact on the 2CH Adaptive filter solution.

The limited signals then enter the first stage 2CH Adaptive filter, which is configured as follows:

Table 4: 2CH MODE First Stage Factory Presets

Parameter	Preset Value
Filter Size	3472 taps
Delay Channel	Input (Primary)
Delay	100 samples
Adapt Rate	16
Adapt Mode	Auto
Output	Residue
Algorithm	LMS
Reference Compensation	Enabled
Crash Detect	Enabled

After the first stage of filtering, the signal enters the second stage 1CH Adaptive filter, which is configured as follows:

Table 5: 2CH MODE Second Stage Factory Presets

Parameter	Preset Value
Filter Size	128 taps
Prediction Span Delay	2 samples
Adapt Rate	32
Adapt Mode	Auto
Output	Residue
Algorithm	LMS
Crash Detect	Enabled

Next, the signal is 200Hz Highpass filtered, to remove any low frequency noises that may remain after processing, and 3200Hz Lowpass filtered to remove high frequency hiss.

Finally, a 10dB AGC is applied to correct for near party / far party talkers, and to provide a good output level for recording.

1CH·ST MODE:

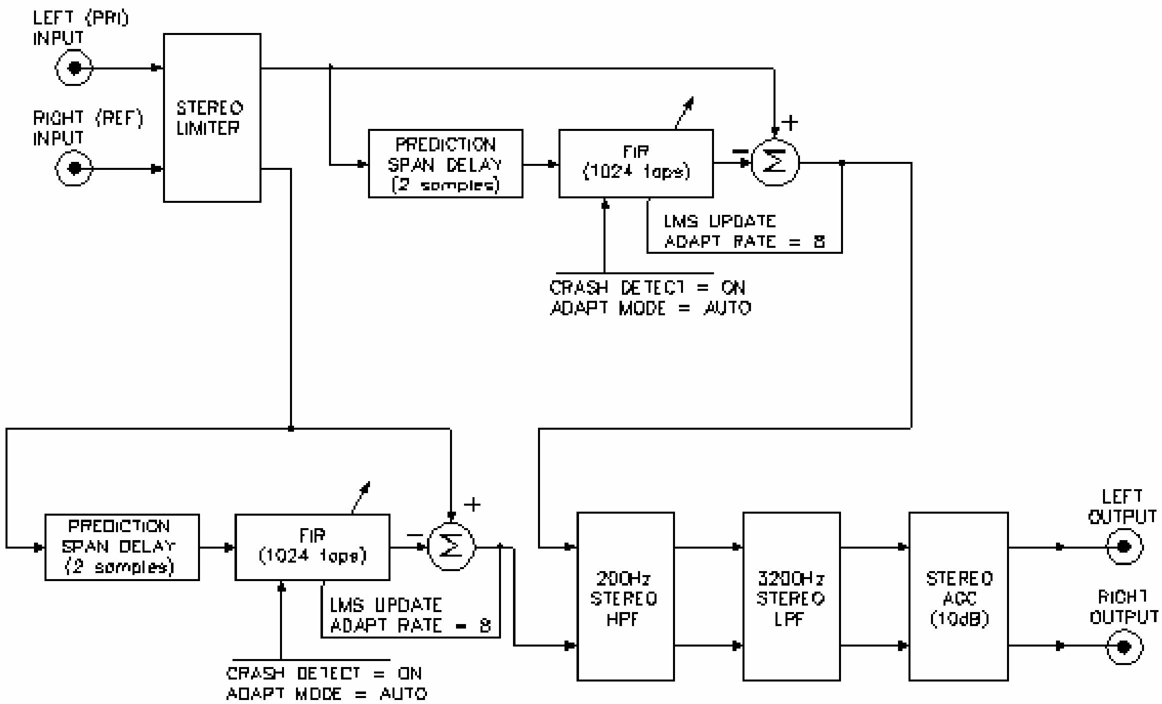


Figure 5: 1CH ST MODE Block Diagram

Notice that in **1CH·ST** MODE, both the **LEFT (PRI)** and **RIGHT (REF)** inputs get fed into the processor. These are usually the stereo output signals from either a stereo microphone preamplifier or a stereo body wire receiver.

The signals are first fed through the stereo **Limiter,** which helps prevent the signals from distorting on overload. The **Limiter** internally "links" the two channels, such that an overload on either input channel will cause the levels to be reduced equally on both channels. This maintains a proper stereo effect on the two signals.

Each limited signal then enters its own 1CH Adaptive filter (two filters total). Each filter is configured as follows:

Table 6: 1CH·ST MODE Factory Presets

Parameter	Preset Value
Filter Size	1024 taps
Prediction Span Delay	2 samples
Adapt Rate	8
Adapt Mode	Auto
Output	Residue
Algorithm	LMS
Crash Detect	Enabled

Next, the signal is 200Hz Highpass filtered, to remove any low frequency noises that may remain after processing, and 3200Hz Lowpass filtered to remove high frequency hiss.

Finally, a 10dB AGC is applied to correct for near party / far party talkers, and to provide a good output level for recording.

3.4 Programmable Filter Descriptions

Filter modes **A-D** are factory preset as large, single 2CH Adaptive filters. Their settings are listed in Tables 7-10. Please refer to the *MicroDAC IV Portable Programmable Digital Filter Configuration Software User's Manual* for complete details on programming filter modes A-D.

Table 7: Programmable MODE A Factory Presets

Parameter	Preset Value
Limiter	Disabled
Filter Size	3600 taps
Delay Channel	Input (Primary)
Delay	100 samples
Adapt Rate	16
Adapt Mode	Auto
Output	Residue
Algorithm	LMS
Reference Compensation	Enabled
Crash Detect	Enabled
200 Hz Highpass Filter	Disabled
Lowpass Filter	Disabled
AGC	Disabled

Table 8: Programmable MODE B Factory Presets

Parameter	Preset Value
Limiter	Enabled
Filter Size	3600 taps
Delay Channel	Input (Primary)
Delay	100 samples
Adapt Rate	16
Adapt Mode	Auto
Output	Residue
Algorithm	LMS
Reference Compensation	Enabled
Crash Detect	Enabled
200 Hz Highpass Filter	Enabled
Lowpass Filter	Disabled
AGC	Disabled

Table 9: Programmable MODE C Factory Presets

Parameter	Preset Value
Limiter	Enabled
Filter Size	3600 taps
Delay Channel	Input (Primary)
Delay	100 samples
Adapt Rate	16
Adapt Mode	Auto
Output	Residue
Algorithm	LMS
Reference Compensation	Enabled
Crash Detect	Enabled
200 Hz Highpass Filter	Enabled
Lowpass Filter	3200 Hz Cutoff
AGC	Disabled

Table 10: Programmable MODE D Factory Presets

Parameter	Preset Value
Limiter	Enabled
Filter Size	3600 taps
Delay Channel	Input (Primary)
Delay	100 samples
Adapt Rate	16
Adapt Mode	Auto
Output	Residue
Algorithm	LMS
Reference Compensation	Enabled
Crash Detect	Enabled
200 Hz Highpass Filter	Enabled
Lowpass Filter	3200 Hz Cutoff
AGC	10dB

5.0 CALIBRATION AND SERVICE

The MicroDAC IV is designed to make calibration unnecessary. However, future firmware upgrades will enhance the present capabilities of the MicroDAC IV. The firmware upgrades will be carried out over an RS-232 link in a similar way to the filter download, using special Windows software to be made available by DAC.

If service is needed, please contact Digital Audio Corporation for assistance.

6.0 MICRODAC IV SPECIFICATIONS

6.1 Analog

Line Inputs	• -29dBm to +14dBm, panel adjustable, Z_{in} = 11k • Rear panel RCA connectors
Line Outputs	• +5dBm, Z_{out} = 50 ohms • Rear panel RCA connectors • S/PDIF Digital Output (44.1kHz)
Phones Output	• Suitable for stereo headset • Front panel volume control and 3.5 mm stereo phones jack
Analog Conversion	• 24-bit sigma-delta A/D and D/A • 12 kHz sample rate • 35Hz to 5.4kHz bandwidth
Level Indication	• Front panel tricolor LED: red (-3 dB), orange (-9 dB), green(-15 dB)

6.2 Digital

Microprocessor	• TMS320C6201 at 1600 MIPS
Computational Precision	• 24 bit signal, 32 bit coefficients, 40 bit accumulation.
Program ROM	• 512k x 8 Non-Volatile Flash Memory

6.3 Construction

Enclosure	• Extruded aluminum with aluminum front and rear panels
Size	• 1.5" H x 7.3" W x 10.0" D overall • 4 lbs
Power	• 9 - 15 VDC at 750 mA • Cable and connector (2.1mm barrel) supplied • AC power adaptor supplied